What is claimed is:

1. An audio signal processing apparatus in which a digital audio signal to be replayed is processed in a waveform thereof, the apparatus comprising:

frequency bandwidth expanding means for expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

low-pass filtering means for performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting means for detecting an interval of time between two waveform peaks of the low-pass-filtered digital audio signal;

difference data calculating means for calculating difference data between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting means for weighting the difference data depending on the interval; and

producing means for producing output data based on the low-pass-filtered digital audio signal and the weighted difference data.

2. The audio signal processing apparatus of claim 1, wherein the detecting means include means for detecting the interval of time two adjacent waveform peaks at which polarities of gradients of the waveform differ from each other.

3. The audio signal processing apparatus of claim2, wherein the interval of time of the two adjacent waveform peaks is measured by the number of times of sampling.

4. The audio signal processing apparatus of claim 2, wherein the past data of the low-pass-filtered digital audio signal used in difference data calculating means are sampled by one sampling period prior to the current data.

5. The audio signal processing apparatus of claim 4, wherein the weighting means is configured so as to weight the difference data depending on the interval and the polarities of the gradients.

6. The audio signal processing apparatus of claim 4, wherein the producing means is configured so as to add the weighted difference data to the low-pass-filtered digital audio signal.

7. A method of processing a digital audio signal to be replayed in a waveform thereof, comprising the steps of:

expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting an interval of time between two waveform peaks of the low-pass-filtered digital audio signal;

calculating difference data between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting the difference data depending on the interval; and

producing output data based on the low-pass-filtered digital

audio signal and the weighted difference data.

8. The processing method of claim 7, wherein the interval of time is an interval of time between two adjacent waveform peaks at which polarities of gradients of the waveform differ from each other.

9. The processing method of claim 8, wherein the past data of the low-pass-filtered digital audio signal are sampled by one sampling period prior to the current data.

10. The processing method of claim 9, wherein the weighting step is configured so as to weight the difference data depending on the interval and the polarities of the gradients.

11. A computer-readable program used for processing an inputted digital audio signal to be replayed in a waveform thereof, the program comprising the steps of:

expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting an interval of time between two waveform peaks of the low-pass-filtered digital audio signal;

calculating difference data between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting the difference data depending on the interval; and

producing output data based on the low-pass-filtered digital audio signal and the weighted difference data.